

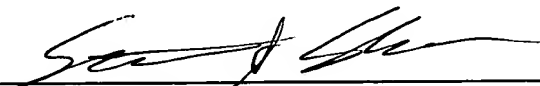
### REMARKS

The text was amended to remove clerical errors. The term in the art "teacher" with the term in the art "referee" to be more accurate as to the supervision required for the automaton. Since it is self-learning without supervision, referee is the more accurate term. No new matter has been added.

New claims 2 through 6 have been added.

Respectfully submitted,

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**VERSION WITH MARKINGS TO SHOW CHANGES MADE**

Patent Application Papers of Clive TANG

behavior in a time-varying Radio Frequency (RF) channel. Voice may only accept a latency of up to about 100 msec,; however, data may bear a much larger value. Voice transmission also requires a certain minimum signal-to-noise (SNR) ratio to be met - a good channel quality would not necessary improve the speech quality, but a poor channel may cause serious deterioration. On the other hand, data is more flexible. Data flow may be increased in good channels to boost the throughput, and conversely, it may be reduced in poor conditions in exchange for a lower bit error rate (BER).

Capitalized on these differences the idea of link adaptation or adaptive modulation, which is the technique adopted in Enhanced Data for GSM Evolution (EDGE) to push the maximum data rate to beyond 384 kbit/s, has emerged recently. In this concept the modulation constellation, coding scheme, transmitter power, transmission rate, and the like, are adapted to the fading channel quality. When the channel is good, a high order modulation with little or no coding is used, conversely when the channel is bad a low order robust modulation is chosen. Several camps of academic researchers have contributed to this subject. Via theoretical and simulation studies, they showed that data throughput and system capacity may be improved or optimized while maintaining an acceptable bit error performance.

Typically, the channel quality is assessed by the instantaneous signal-to-noise (SNR) ratio, which is divided into a number of fading regions, with each region mapping into a particular modulation scheme. Thus one basic issue in adaptive modulation is to determine the region boundaries or switching thresholds, i.e. *when* to switch between different modulation schemes. A common method [is to set] for setting the thresholds to the signal-to-noise ratio (SNR) required to achieve the target Bit Error Rate (BER) for the specific modulation scheme under additive white Gaussian noise (AWGN) has been shown in the art. While this maintains a target BER, this does not optimize the data throughput which is probably a more important concern for data transmission. In Nokia's (Finland and Irving, Texas) joint "1XTREME proposal" with other companies to 3GPP2, the switching thresholds are derived from steady state throughput curves of the individual modulation schemes. This increases the through[t]put relative to the previous method but still is not optimal. For packet data transmission in a time-varying channel, what would be desirable is an on-line adaptive scheme that can adjust the switching thresholds dynamically to maximize the throughput.

## A BRIEF DESCRIPTION OF THE DRAWINGS

The above set forth and other features of the invention are made more apparent in the ensuing Detailed Description of the Invention when read in conjunction with the attached Drawings, wherein:

Figure 1 shows block diagram of the test system[ is shown in];

5       Figure 2 is graph showing of BER vs SNR;

Figure 3 shows a graph of the switching thresholds that are derived from steady state throughput curves of the individual modulation schemes;

Figure 4 shows a block diagram of an automaton/environment model; and

10       [.]Figure 5 shows the probability convergence curves of desired action for SNR of -  
1,0, and 1 dB.

## DETAIL DESCRIPTION OF THE INVENTION

The present application provides for a scheme for an on-line adaptive scheme that can adjust the switching thresholds dynamically to maximize the throughput. We first set up a simulation system comprising of selectable, convolution encoded QPSK, 16QAM and 64 QAM sources, a flat Rayleigh fading channel model, coherent demodulators and soft Viterbi decoders. By means of this test bed, the effect of altering the switching thresholds on the data throughput can be revealed. It will be shown that a significant increase in throughput may be obtained by merely altering the value of one threshold. Next, an on-line adaptive learning scheme will be introduced that is capable of adaptively optimizing the switching thresholds as the data is transmitted. A key feature of this self-learning scheme is that it does *not* require a dedicated training signal, instead it utilizes the long-term throughput as the [teacher]referee to train up the learning algorithm. The scheme will be demonstrated to converge to the best threshold value available that maximizes the long-term average throughput.

## SYSTEM MODEL

To study the application of novel learning schemes, we start with a simple system model and operating scenario. A straightforward system configuration with basic settings is preferred as the current aim is to explore new ideas and novel concepts. We assume that the modulation scheme selection in the transmitter is reliably passed on to the receiver so that the data may be properly demodulated. We also suppose that information regarding failure frames is available to the transmitter (e.g. a single bit from the receiver to indicate whether or not the transmitted frame passes the CRC). In a practical system, these may be implemented by reserving extra slot spaces in both forward and reverse links. Furthermore, we assume perfect channel estimates are available so that coherent demodulation may be performed.

A block diagram of the test system is shown in Figure 1. A random source 110 is used to generate a stream of binary digits, from which 184 bits are taken at a time and 8 flush bits added to form a frame. The created frame is then encoded by use of a convolution encoder 120 with constraint length  $K=9$  and a rate  $R=1/2$ . (The frame structure and generator polynomial are taken from the latest cdma2000 standard as an example. Those skilled in the art after reading the specifications may arrive at variations which are deemed to be in the spirit and scope of the invention). One frame of data thus corresponds to 384 encoded bits. Three different schemes 130 are available to modulate the encoded

bits - QPSK, 16QAM and 64QAM which takes in 2, 4 or 6 encoded bits respectively at a time to create a modulated symbol. A modulated frame, which comprises of 192 modulated symbols, therefore consists of either 1, 2 or 3 frames of data. For a given modulated symbol rate  $x$ , the frame rate  $y$  is thus equal to  $x/192$  resulting in a data rate varying from 184y to 552y.

The channel model used is a single path flat slow Rayleigh fading channel 150 with the Doppler frequency set to 5 Hz. Because the channel fades slowly, the channel is only monitored once per frame, at the beginning of the frame. The appropriate modulation scheme is chosen based on the measured instantaneous SNR, with the scheme maintained for the entire frame of data. That is, the modulation scheme is only allowed to vary on a frame-by-frame basis.

At the receiver, the symbols are coherently demodulated 160 and soft Viterbi decoded 170 to recover the original data. One frame of demodulated symbols are decoded at a time, producing 1, 2 or 3 frames of data depending on the modulation scheme used. Frame error information is fed back to the transmitter 140.

In the present application, the transmitted power level and the coding rate are kept constant, we only focus on adapting the data transmission rate by varying the modulation scheme according to the measured SNR. When the channel condition is very bad, no data transmission takes place. Hence it is a modulation-level-controlled adaptive modulation, in a similar manner as described in art.

In addition to BER, the performance of the adaptive modulation system may be assessed by the long-term Frame-Error-Rate (FER), defined as the ratio of the number of corrupted frames to the total number of data frames transmitted; and the normalized long-term average throughput TP, defined as  $TP = (1-FER)*FPB$ , where FPB is the average frames-per-burst that varies from 1 to 3. The maximum value of TP is 3, when data is transmitted with 64QAM and no frames are received in error (i.e.  $FPB=3$  and  $FER=0$ ). The minimum value is 0, when all frames are corrupted or no transmission occurs (i.e.  $FPB=0$  or  $FER=1$ ).

#### DETERMINATION OF SWITCHING THRESHOLDS

In a modulation-level-controlled adaptive modulation the key parameters are the switching thresholds that determine when to switch from one modulation scheme to

another. In the present system, that employs three modulation schemes. There are three switching thresholds to be determined - from no transmission to QPSK (threshold L1), from QPSK to 16QAM (threshold L2), and from 16QAM to 64QAM (threshold L3). One approach is to set the thresholds as the SNR required to achieve a certain target BER for the specific modulation scheme under AWGN. By first plotting a set of BER vs SNR graphs as depicted in Figure 2, and then setting a target BER the switching thresholds L1, L2 and L3 may be read directly from the graph. For instance, for a target BER of 0.01, L1, L2 and L3 may be set to 1.4, 6.6 and 10.8 dB respectively as indicated by the dotted lines. This setting maintains the target BER, however it does not optimize the data throughput. Torrance and Hanzo also suggested a numerical optimization method [9], but it requires the throughput to be obtainable as an analytical function of the thresholds which is generally unavailable in a practical system.

In Nokia's joint 1XTREME proposal to 3GPP2, the switching thresholds are derived from steady state throughput curves of the individual modulation schemes. Figure 3 shows such a graph for the test system. The idea is to use the modulation scheme that gives the best throughput for the given SNR. The switching thresholds are suggested by the dotted lines, but the graph does not tell when to turn on from no transmission to QPSK (threshold L1). This method may increase the throughput relative to the previous one, however it is still not optimal.

Simulations in the test system quickly revealed that the average BER, FER and TP can vary a lot by altering the switching thresholds. This, coupled with the time-varying nature of a RF channel, suggests what would be desired is an *on-line* adaptive scheme that tailors the switching thresholds *dynamically* to maximize the throughput (or other chosen criteria) as the data is transmitted. Furthermore, because of the difficulties in deriving TP as an analytical function of the switching thresholds in practical situations, it would be advantageous to use a *self-learning* method that does not utilize expressions of TP and the thresholds, nor makes any assumption of the operating environment. The scheme should be able to carry out *global* optimization in case the performance criterion is a multi-modal function. Equally important is that it should be easily implemented in a mobile transceiver. It would also be attractive *not* to use any dedicated training sequence in order to reduce the overhead. A class of adaptive learning techniques, namely stochastic learning automata, fits in this description and is hereby proposed as the modulation selector.

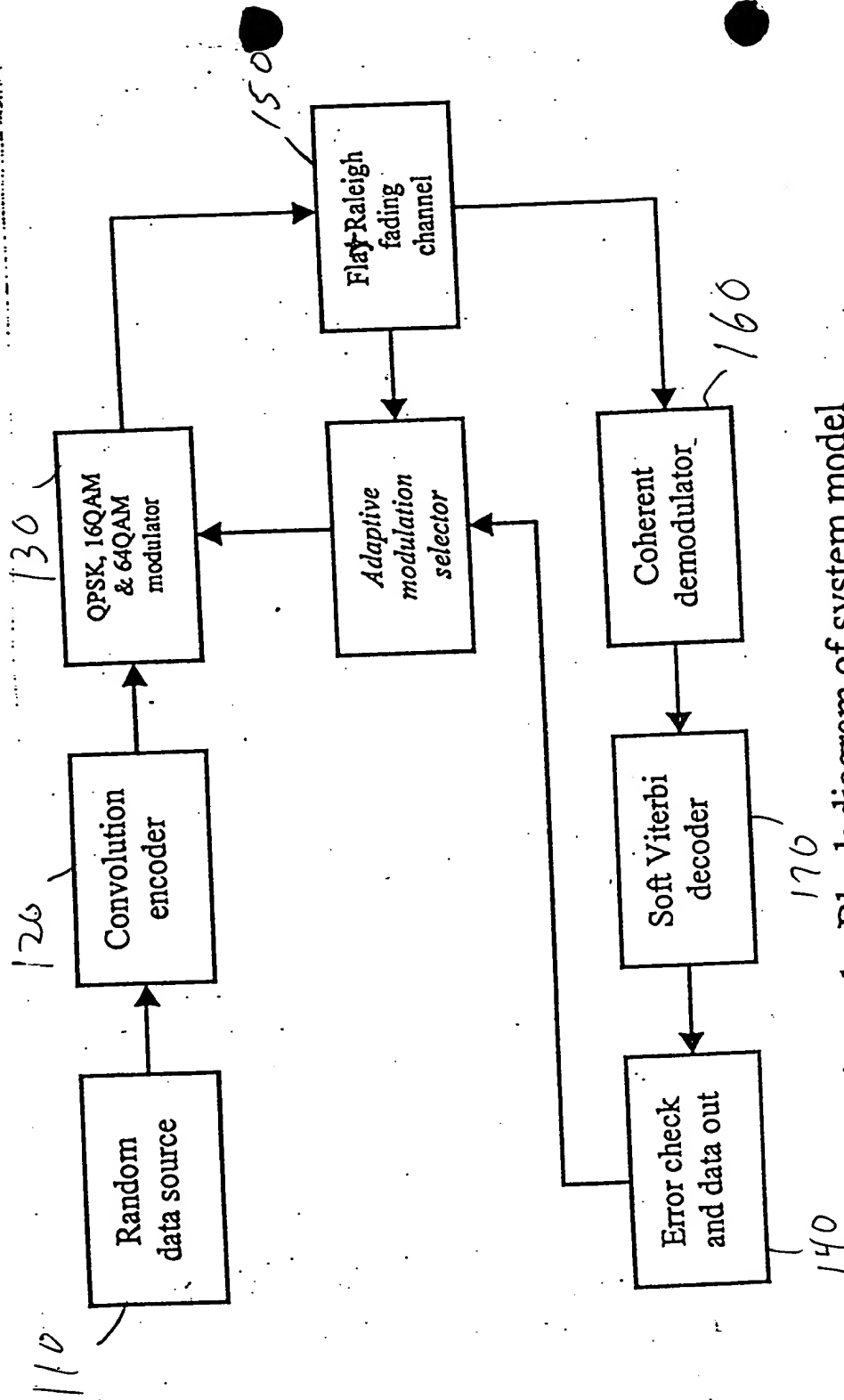


Figure 1 - Block diagram of system model



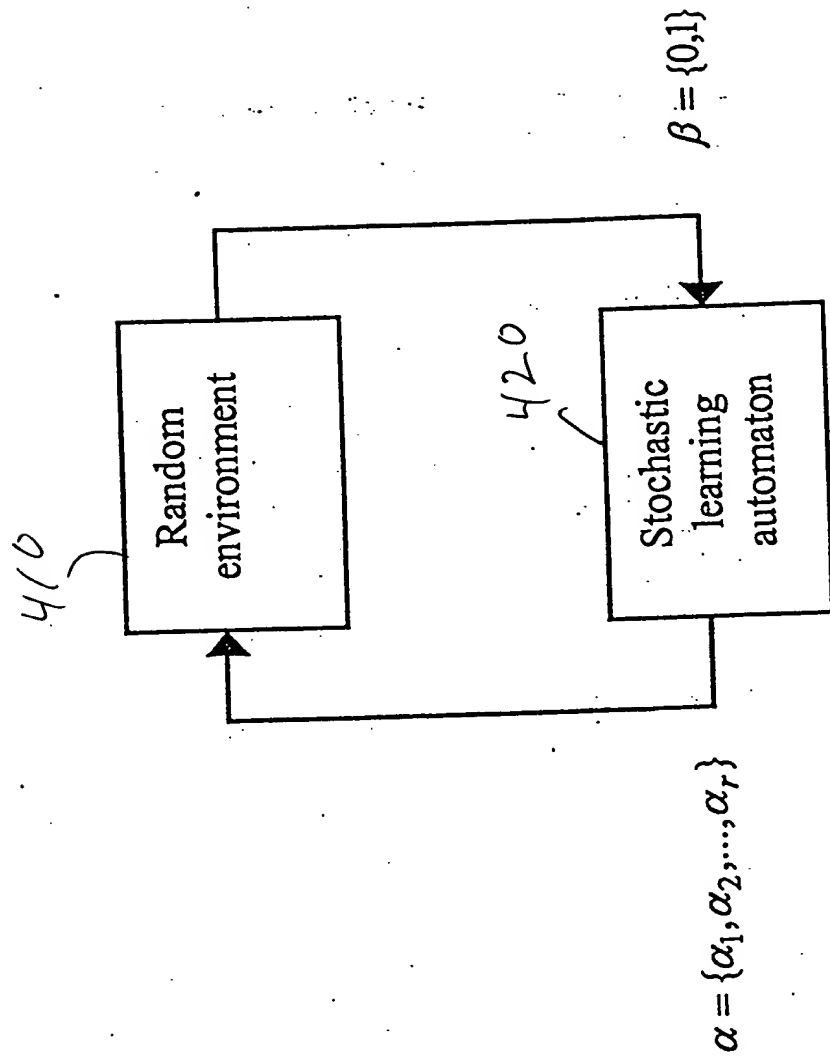


Figure 4 - Stochastic learning automaton operating in a random environment